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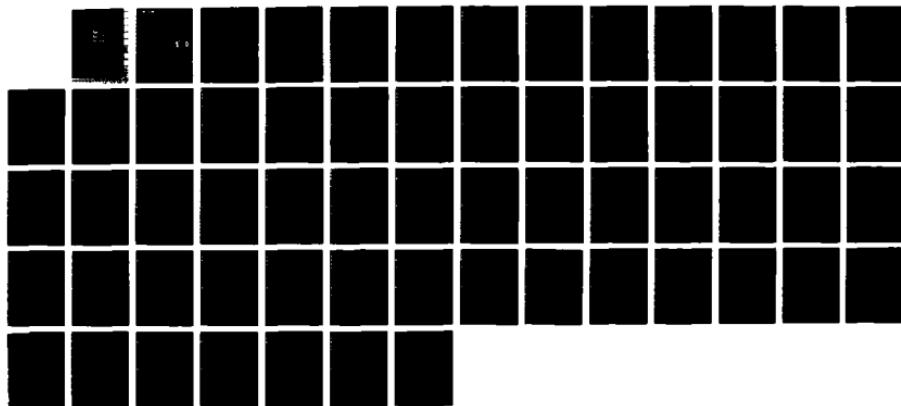
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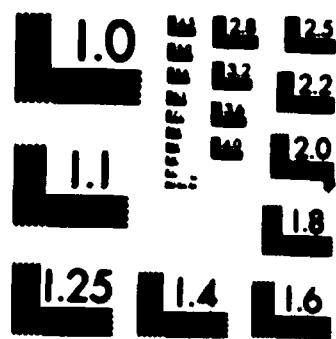
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MOBILE COMPUTER NETWORKS

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28 APRIL 1987

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probability and relative measure of congestion.

MOBILE COMPUTER NETWORKS

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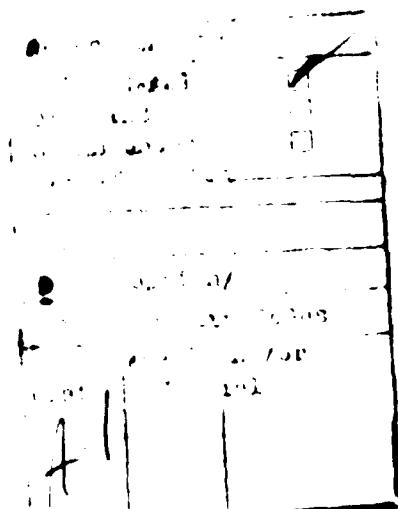
Steven John Janis



**A Thesis Presented in Partial Fulfillment
of the Requirements for the Degree
Master of Science**

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May 1987



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ABSTRACT

The current cellular system provides a predetermined number of mobile subscribers dedicated telephonic service if they are within transceiver range of its backbone network. The subscriber's mobility is limited to the coverage area of the cell grid and is further bound by the number of dedicated channels available in a cell. This report proposes a mobile computer network wherein the terminals double as relays affording the mobile subscriber greater mobility and where all terminals share a common transmission channel.

The main design issues of the proposed mobile computer network's data link and network layers are investigated. The network incorporates a nonpersistent Busy-Tone multiple access channel as the means of linking all stations together. A unique routing algorithm which maximizes packet discarding and maps the network topology is described and used. The congestion control techniques incorporated in this network reduce even further the networks blocking probability and relative measure of congestion.

Keywords: Thesis; packet radio networks

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LIST OF TERMS

ACK (Acknowledgement). A field in a packet or a special packet that notifies the sender that the transmitted data was correctly received.

ALOHA. A channel access control technique for multiple access transmission mediums. A station transmits whenever it has data to send. Unacknowledged transmissions are repeated until received.

Backbone. The fixed interconnection of stations or nodes in a network.

Blocking. The inability of a station to either seize the channel for the purpose of transmitting data or receive data because its queue is full.

BTMA (Busy Tone Multiple Access). A channel access scheme that assists in the prevention of packet collisions through the use of a secondary common channel that notifies other stations that the primary channel is occupied.

Capture. The ability of a receiver to detect the strongest overlapping packet and receive it correctly.

Congestion. The condition when there are input traffic rate in (a part of) the network exceeds the service rate causing undue time delays.

CSMA (Carrier Sense Multiple Access). A channel access control technique for multiple access transmission mediums. A station desiring to transmit data first senses the channel and only transmits its data if the channel is idle.

Cycling. See looping.

Data Link Layer. Layer 2 of the OSI model which provides for the reliable transfer of information across the physical link. It creates and recognizes frame boundaries, processes acknowledgement frames and controls error handling.

Deadlock. An advanced condition of congestion where no station can forward packets because there are no buffers available.

Decentralized routing. Each station is responsible for the routing of all packets that it encounters.

End-to-end control. A monitoring system where the packet's rate of flow is limited at the originating station prior to entering the network.

External customer. A customer from the local station that generates messages which go into his queue and are serviced.

FCC (Federal Communications Commission). An agency of the federal government which regulates all means of communication, and is also responsible for the assignment of all broadcasting frequencies.

Field. A portion of a frame, measured in bits, that contains either control or data information.

Frame. A bit string that contains both data and control information.

Global. A term used to describe an algorithm that can be applied to all stations in the network and yield the same results.

Half duplex. A data transmission in either direction, but not simultaneously.

Handoff. The transfer of control from radio to radio in the cellular system.

Header. System control information that precedes the data in a frame.

LCO (Local Central Office). The switching center that connects an MTSO to the fixed landline network.

Local control. A monitoring system where restrictions on the packets are applied once the packets are in the network.

Local customer. A customer who originates a message from the first queue in the tandem link model.

Looping. The process of having traffic rerouted through the same node more than once.

Macroscopic diversity. A term used to describe large scale signal reception compensations resulting from large obstacles and/or large terrain deviations in the communication path.

Microscopic diversity. A term used to describe small scale signal reception compensations on the order of multi-path fading.

MTSO (Mobile-Telephone Switching Office). It provides call supervision and control for all mobile calls and acts as the switch interface between the mobile and fixed landline networks.

Network Layer. The third layer of the ISO model which is responsible for packet routing, congestion control and accounting information.

Nonpersistent CSMA. A medium access control scheme that senses the channel and transmits only if the medium is not busy. If the channel is busy it waits an amount of time determined by a programmed probability distribution and then senses the medium again in an effort to transmit the message.

OSI (Open Systems Interconnection). A model developed by the International Organization for Standardization that describes the seven standardized layers of computer communication architecture.

Packet. Quantized data that is sequenced which recombines into the original message.

Protocol. The set of rules or conventions controlling the exchange of data between two stations; where the key elements are syntax, semantics, and timing.

Routing. The process of transmitting data in the most direct route through a network while trying to minimize congestion and avoid deadlock.

Slotted ALOHA. A channel access control technique for a multiple access medium where a station having data to transmit can only do so at the beginning of a synchronized time slot. Unacknowledged transmissions are repeated as above.

Throughput. A measure of channel utilization or capacity resulting from specified parameters of offered load and/or delay times.

CHAPTER 1

INTRODUCTION

Today, the term "mobile-radio communications" describes any radio communication linked system between two terminals at unspecified locations within a given region where one or both are either moving or not. This implies that one may actually be at a fixed location, such as a base station. Therefore, this term applies to both mobile-to-mobile and mobile-to-fixed radio communication links; which also describes the current cellular radiotelephone systems [14].

Although the current cellular systems are considered mobile they still require a backbone system for support; thereby, restricting the mobility of these terminals to within certain established regions. If current cellular radiotelephone systems correspond to the first generation of mobile-radio communications, then the next generation should employ a truly mobile system where all terminals are independent of the backbone (fixed transponders and switching) support system. This next generation, mobile-radio communication system, is the focus of this paper.

The focus of this investigation will be placed on the Network and Data Link Layers of the seven layer Open Systems Interconnection (OSI) reference model (when packet broadcasting is used normally these layers

of the OSI model are combined into a single layer which will be termed the Network Layer). Network access, routing and congestion control will be the main topics covered.

A packet radio network with a fluid topology is the underlying objective of this paper. Due to the nature of the network under investigation only decentralized routing will be considered. This routing idea allows data to originate from any node in the network and is forwarded by any other node in the network within the transmitting radius of the originator node to the destination node. This type of a routing algorithm yields a highly robust means of routing traffic within the network. This mobile communication system has been chosen to be totally independent of the current telephone system because of the fluid topology requirement. This requirement does not preclude the network from being connected to the current telephone system with the use of gateway devices. It also has the potential to handle new clients that are not yet connected to the current telephone system; and can accommodate sudden growth; and can easily provide service to undeveloped areas in need of a communication system.

CHAPTER 2

BACKGROUND

2.1 GENERAL CELLULAR DESCRIPTION

Currently mobile stations meander through a lattice of fixed transceivers which are connected to the main telephone system, where all routing decisions are made. This system has an inherent backbone structure that provides services to its mobile terminals. The existing commercial fixed telephone network overlies the cellular network and the primary interface is via the mobile-telephone switching office (MTSO) facilities. The lattice is comprised of a number of hexagonal cells having radio transceivers with directional antennas located at alternate corners in these cells (as shown in Fig. 1). These cell boundaries are defined by the minimum required signal strength at distances determined by the reception threshold limits.

The cellular mobile-radiotelephone system can be expected to accommodate the growth of new subscribers in two ways. First, not all of the channels allocated to a cell are initially placed into service. As the numbers of mobile subscribers and the traffic intensity increases, transmission facilities for the additional channels are modularly expanded to keep pace with the demand. Second, as the number of channels

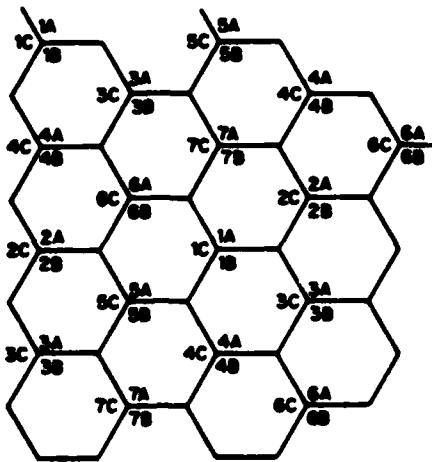


Fig. 1. Cell plan with frequency set number.

per cell approaches the maximum within the channel allocation plan, the area of individual cells can be reduced, thus permitting more cells to be created with less physical separation but with increased reuse of assigned channel frequencies [14]. This reconfiguration of the cellular network permits the same number of assigned channels to adequately serve a larger number of mobile units within a greater number of smaller cells. The ideal, customized cellular network would not be uniformly divided into cells of equal size but would contain cells of different sizes based on the density of mobile units within the various cell coverage areas.

Since this system design utilizes the reuse of Federal Communications Commission (FCC) allocated frequencies, the number of customers serviced is greatly increased.

This system has several advantages:

1. Direct-dialing features equivalent to those offered to fixed-telephone subscribers.
2. Absolute privacy of communication, with greatly improved quality.
3. An extended range of communication utilizing the total switching resources of the commercial telephone networks.
4. A theoretically unlimited number of communication channels that can be provided. (Above 1GHz, atmospheric conditions such as moisture and climatic effects must be taken into consideration. These effects are minimal at operating frequencies below 1 GHz. Below 30 MHz, path loss and signal fading are not severe.)

Control is transferred to the antenna providing the strongest average signal from a mobile terminal during any given time interval. Furthermore, periodic analysis of channels in use would determine the necessity for handoff to an alternate cell within the primary cell area (intrahandoff), or handoff to a cell in an adjacent area (interhandoff). All of these decisions would be made automatically without the knowledge or intervention of the user, and without interruption of the call in progress.

2.2 CELLULAR ACCESS

A call originating from or terminating at a mobile unit is serviced by a cell site connected via landlines to a mobile-telephone switching office (MTSO). The MTSO provides call supervision and control, and extends call access to a commercial telephone landline network via a local central office (LCO) telephone exchange, a toll office, and any number of tandem offices required to complete the call path. The terminating central office completes the connection to the called subscriber at the distant location. Two types of mobile-radio channels are used in setting up a call: paging channels and communication channels. The mobile unit is designed to automatically tune to the strongest paging channel in its local area for continuous monitoring, and to automatically switch to another paging channel when approaching the threshold transition level of reception.

2.3 CELLULAR MOBILE TO MOBILE ROUTING

The mobile unit's telephone in going off hook signals a request for service over a paging channel chosen by its receiver. After the MTSO identifies the location of the serving cell site, an idle communication channel is assigned to the mobile unit via that cell site and a dial tone is returned to the mobile user. The MTSO will transfer the call to its LCO where it then enters the telephone system. The telephone switching network translates the called number and routes the call to the MTSO in the cell area closest to the called mobile-telephone subscriber. The terminating MTSO determines whether the number called is busy or

available. If busy, the MTSO causes a busy signal to be returned to the calling party. If the mobile subscriber's telephone is available, the called number is broadcasted over all paging channels assigned to cell sites in that area. The mobile unit automatically recognizes its number and responds by an acknowledgement (ACK) over the corresponding paging-channel frequency. On the basis of the paging channel response, the MTSO will identify the serving cell site and automatically switch the mobile unit to an idle communication channel from among the channels allocated to that serving cell site. The MTSO, after selecting an idle communication channel, causes the mobile unit to tune to that channel by means of a data command over it. The incoming call is connected to the appropriate circuit serving the mobile unit, and a ringing signal is sent to the mobile unit.

Each cell site has a locating receiver which monitors all active channels in discrete time intervals. After having served as the central office in completing the call setup, the MTSO continues to access the serving cell's locating receiver, thereby monitoring the mobile-radio transmissions at prescribed intervals. Should the received average signal strength drop below the prescribed level in any given time interval, the MTSO will automatically and without interruption, switch the call to an idle channel in a cell site serving the mobile unit which has the strongest received signal above the prescribed level. This is the handoff process, and it can be performed within the same cell or in a new cell.

2.4 CELLULAR CONGESTION/BLOCKING

Earlier it was mentioned that the cellular network has a limited number of FCC authorized frequencies so frequency reuse is a key issue in these networks. Normally, ten to twenty channels are allocated to a standard cell. If all the channels are occupied the call, be it an original or handoff, is blocked. This blocking is considered call congestion in the cellular vernacular. When the blocking probability reaches a predetermined level, cell splitting is performed to contain this congestion.

Under normal design conditions, in an evenly distributed traffic pattern, about five percent blocking can be expected in a single cell if the system is properly designed. If a directed retry approach [4] is utilized the blocking probability is lowered to roughly two and one half percent.

2.5 CELLULAR AND PROPOSED NETWORK COMPARISON

The current cellular system utilizes the omnidirectional radiotelephones in mobile units that must be within a specified range of one of the directional relays in the fixed cellular grid. These mobile stations access the network via a paging channel and obtain a circuit switched voice channel from the MTSO. The mobile station simply dials a telephone number and the call is routed and switched via the landline telephone system.

The proposed mobile system requires the use of omnidirectional packet radio-computer units in its mobile stations. Each transmission is initiated by a mobile unit first accessing a common signaling channel and then transmitting a packet specifically addressed for a destination to its neighboring mobile unit(s) which then relay the packet using the same procedure.

Table I references the major points of interest between the cellular network and the proposed network.

Table I
Cellular and Proposed Network Comparison

Item	Cellular	Proposed
Terminal mobility	Yes	Yes
Communication implement	Telephone	Computer
Radio configuration	Voice	Packet
Radio channel (medium)	Dedicated	Common
Terminal antennas	Omni	Omni
Backbone structure	Hex lattice	None
Relays/switches	Fixed	Mobile
Type of switching	Circuit	Packet
Routing means	Direct-dial	Address
Access medium	Paging ch	Common ch
Relay antennas	Uni	Omni
Routing decisions made	Switches	Terminals

CHAPTER 3

MODEL AND ASSUMPTIONS

The cornerstone of this mobile computer network model is that it utilizes a packet radio system, which is restricted to a half duplex mode of operation. The packet radios will employ omnidirectional antennas to afford uniform circular area coverage. Furthermore, all radios transmit with the same output power yielding the same radial distance areas of coverage. To avoid macroscopic diversity considerations it is assumed that there are no obstacles or large deviations in the terrain to interfere with our signal. In avoiding microscopic diversity concerns all mediums are assumed to be noise free, and there is no multipath fading or use of capture ratios [3]. It is further assumed that the propagation time of the carrier and the time required to detect it are negligible; therefore, small nonzero propagation and detection times are assumed. Also the packet radio will operate in the same frequency band as the current cellular system [2], [6].

The packets are assumed to be a fixed length. Arriving packets at a node are assumed to occur according to independent Poisson processes. The average time required to transmit a packet (service rate) is $1/U$ sec; and all nodes have the same expected load and mean arrival rate for an evenly distributed traffic pattern. Initially unlimited buffer space is

assumed; but will become fixed during the investigation of congestion control.

Finally, since very little is known about the movement behavior pattern of mobile terminals, it is assumed that under the worst conditions a terminal can communicate with at least one other terminal. Obviously, to be completely disconnected from the network is possible, but yields no valuable insight into the proposed network. It should also be mentioned that it is possible to become completely disconnected in the cellular network by merely moving out of the network's coverage area.

CHAPTER 4

CHANNEL ACCESS

Packet radio requires a form of multiple access control that permits many radios scattered about a specified region to transmit on the same radio frequency without interfering with each other's transmissions [9], [12], [13]. Restricted to a half duplex medium implies that the device can either transmit or receive packets on the common radio channel during a time interval. Then, for a station to receive a packet, it is required that the radio does not transmit during that same period of time. Also, to preclude any packet collisions, a further requirement is that no neighbor of either the transmitting or receiving stations be allowed to transmit (shown in Fig. 2).

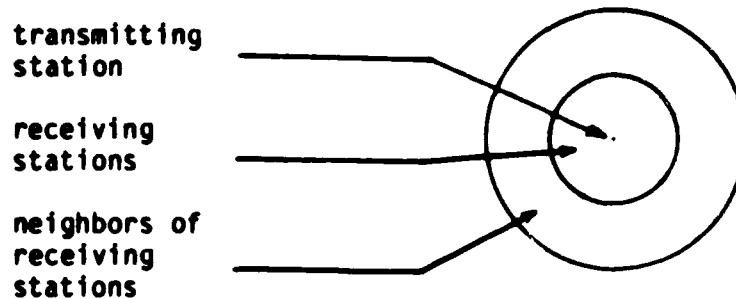


Fig. 2. Inhibit zone for transmissions.

Implementation of this concept requires that each packet radio device utilize two different radio channels, and is intelligent in order that it be able to determine whether or not it is allowed to transmit.

The protocol is as follows: a station desiring to transmit does so on the common radio channel. All nontransmitting stations monitor this common channel and upon sensing activity emit a signal (of the same transmission power) on another common channel (a busy-tone channel). Upon detecting a signal on the busy-tone channel no station would try to transmit until this busy-tone channel was clear. This mechanism will inhibit all the neighbors of the transmitting station and the neighbors of the receiving stations permitting a successful transmission.

Packet radio networks are normally associated with either the ALOHA access [10] scheme given as (with S being the throughput and G being the offered channel traffic rate)

$$S = Ge^{-2G}$$

or SLOTTED ALOHA access scheme given as

$$S = Ge^{-G}$$

Kleinrock and Tobagi proved that nonpersistent carrier sense multiple access (CSMA) given as (where a is the ratio of propagation delay packet transmission time) [21]

$$S = \frac{Ge^{-aG}}{G(1+2a) + e^{-aG}}$$

provides better throughput-delay (G/S) characteristics than does either ALOHA scheme (see Fig. 3) [11], [19], [23]. Furthermore, they found that in the mobile realm nonpersistent CSMA's throughput performance was severely degraded (from a maximum of .815 to a maximum of .44) by the addition of mobile units that were not within sensing range of a transmitting station (between R and $2R$ as shown in Fig. 2). Since these stations were not able to sense any traffic on the channel they would transmit packets that would collide with the other station's transmission (between 0 and R as shown in Fig. 2).

Tobagi and Kleinrock determined [23] that the upper bound on throughput of a nonpersistent BTMA is given as

$$S_u = \frac{b_m f e^{-(g T_m)} + (1-f) e^{-gm(0, T_m)}}{W B_L + I}$$

where: b_m is the number of bits per packet; W is the channel bandwidth; g is the rate of the offered channel traffic; T is the transmission time of a packet; f is the fraction of the time gap between busy signals; and B_L and I are determined from the following equations:

$$B = T_m + \frac{e}{\Delta g} \Delta g T_m [1 - e^{-g \Delta T_m}] - \frac{-g \Delta T_m}{\Delta g e} (f + T_m - f T_m)$$

$$I = \int_0^\infty e^{-gm'(0,z)} dz$$

They also found that the closer f gets to zero the more accurate the lower bound becomes. The lower bound is given by

$$S_L = \frac{b_m}{W(B+1)} e^{-gm(0,T_m)}$$

Therefore, the implementation of the busy tone multiple access (BTMA) method greatly reduced this interference of the other terminals thereby creating only a minimal degradation in the throughput-delay characteristics of the nonpersistent CSMA protocol (see Fig. 3).

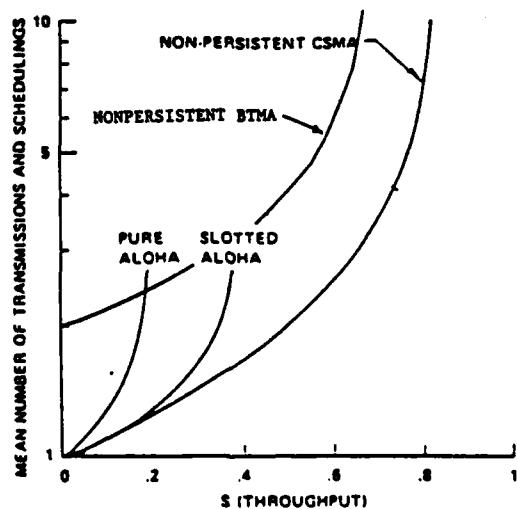


Fig. 3. G/S versus throughput.

CHAPTER 5

ROUTING

Once the channel has been successfully seized, only then does routing become a primary issue. Routing is the process of selecting the best path for traffic flow in a network. Therefore, the scope of the routing algorithm should be designed to primarily handle the average trends in traffic flow. This requires that the routing algorithm be global and dynamic [16] in nature (highly robust) and provide the best possible service to the network. A highly robust routing algorithm will not necessarily prevent congestion but it will inevitably assist in its control.

This packet radio system having numerous terminals that double as repeaters, and are allowed unrestricted mobility, present some unique problems utilizing the current routing strategies. One, obviously, is that it would be a rigorous and costly task to know and track each radio in some hierarchical fashion and another is that of packet proliferation.

A suitable working algorithm that accomplishes this difficult task requires the use of three special fields in the packet's header:

1. a hop count field
2. a route field
3. a change field

The hop count field is used to limit packet proliferation by placing an upper bound on how many times the packet will be transmitted within the system. Every time the packet is successfully transmitted and received the hop count is decremented by one; when the hop count reaches zero the packet is discarded. To ensure the packet arrives at its destination the hop count must be sufficiently large to accommodate any network topology. Then by selecting the hop count, H , such that it equals the maximum number of stations, N , in a one-connected network ($H=N$; see Fig. 4) the packet is guaranteed to reach its destination.

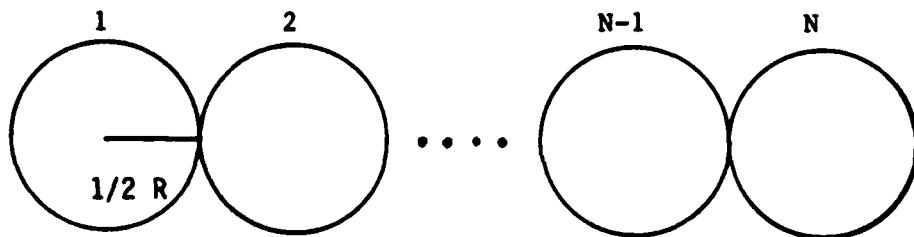


Fig. 4. Worst case, one connected network.

The route field is used to provide routing and handling instructions for the other stations in the network. This field must then contain H "instructions". An "instruction" is understood to be the address of the next station(s) enroute to the destination, of which there are three basic types:

1. ALL meaning all stations are to receive this packet
2. NONE meaning the packet is not to be forwarded
3. SOLE intended for a particular station

During routing the stations scan the route field and if their SOLE address is in any position other than the receiving one (same as the hop count), the packet is discarded. It is further taken that the stations have memory and can determine from the route field (by comparing previous addresses, packet number and destination) if they had heard the packet before. If they had heard it before, the packet is then discarded. Therefore, this algorithm has three cases in which packets can be discarded:

1. hop count going to zero
2. having NONE as the next instruction in the route field
3. having handled the same packet twice.

The change field is used to show how many times the established route has been modified during a transmission. If the route field is changed more than a fixed percent during a transmission, a new route will be established.

The first packet of any message is always transmitted with every instruction in the route field set to ALL. As this packet proceeds through the network each station handling it replaces the specific ALL corresponding to the number of hops made with their (SOLE) address; the hop count is then decremented and the packet retransmitted to all stations per the next route field instruction. The destination station upon receiving this first packet generates an end-to-end ACK, which includes a copy of the packet's route field (remaining instructions set

to NONE), and transmits it to the originating station along the reverse path of the route field. This path now becomes the primary route for the remaining packets in the message destined for that terminal. If during a transmission (after the path has been determined) either a station is blocked or has moved out of range only that SOLE address will be replaced by an ALL, the change field will be incremented by one, and the packet will be retransmitted. The receiving stations of this ALL transmission place their SOLE address in the ALL position and transmit the packet to the next SOLE address. If the moving or blocked station is the destination, then its SOLE address' location will be incremented by one and an ALL will replace its previous place. If the change field exceeded its predetermined threshold the destination transmits another end-to-end ACK adjusting the route to the new topology.

The advent of the route field provides three useful purposes. First, stations are prevented from handling the same message twice. Second, looping or cycling is prevented from occurring. Third, this field provides a general topology of the network for routing the remaining packets in the message. The route field also affords a packet the ability to get within close proximity of a terminal and "track" it if the terminal moves into an adjacent cell (insuring that the hop count constraint does not go to zero).

This judicious use of the hop count coupled with the route field permits only the first outbound packet to experience any type of controlled proliferation. An end-to-end ACK is to be used with either

the initial packet or whenever the route field has been changed more than threshold limit. Throughout the routing process it is assumed that a station can hear the other stations retransmitting its packet and therefore verify its correctness. This precludes the need of any hop-to-hop ACK.

At this point in time an investigation of the frame size is of the utmost importance. The frame cannot be too large or too small. If the frame is too small, a large percentage of it is header information and it would require that many frames be transmitted to send a message. If the frame is too large, bandwidth is wasted if the message to be transmitted is small compared to the relative frame size. Therefore, a median frame size must be determined. The addition of the hop count, route and change fields makes the header the primary concern at this point. It is conceivable that the route field can take on quite enormous proportions if the network were very large.

An investigation reveals that there are basically three ways to implement the route field. First, the route field can be given as a string of address bits in the header (H the number of address bits). Second, the route permutations of the network can be placed in a stack in another memory location and the route field then would be the number of bits required to reference a specific level in the stack. Third, using the results from method two, the stack addresses are placed into a square matrix and the route field is now split to index in both the horizontal and vertical directions.

For example, if a 32 station network were to be established: using the first method, the minimum size of an address would be five bits ($2^5=32$), implying a 160 bit route field; a five bit hop count field; and (if the change threshold is 20%) a three bit change field, for a total of 168 bits. Using the second method, the same 32 station network would now require a 125 bit route field yielding a total of 133 bits. Employing the third method, the 32 station network now would require an 8 bit route field yielding a total of 16 bits. Consequently, the size of the header is directly related to the size of the network. If the frame size is limited in length method three yields the most viable solution.

CHAPTER 6

CONGESTION CONTROL

The control of congestion is an important objective in the design of switched communication networks [8], [18], [20], [22]. Congestion occurs when more traffic enters a network than can reasonably be served, which may be caused by either system failures or peak traffic flow. Of the two underlying causes only the control of peak traffic flow will be investigated.

The usual way in which switched networks are designed is to assume nominal demands for each station, and then specify the network topology, routing strategy, and link capacities to handle that demand with some desired quality of service. If at any particular time a user of the network has an increase in usage needs, that are outside the design specifications, the stations' service from the network will not be as good. In addition, that user can also cause a degradation in the service given to others as a result of his increased demands on the network. This is due to the fact that in switched networks the resources are shared by all the other users in the network and unexpected interferences may/will occur. If limited buffer space is available at intermediate stations, buffers can become filled and congestion can spread throughout the network. At some point, overall network performance can become

unacceptable and the network is said to be congested. In extreme cases, deadlock conditions [7] can occur and outside intervention is needed in order to restore service. Therefore, the two key parameters in measuring congestion are; not being able to access a station or the medium (blocking probability); and time delay.

There are three major classes of congestion control measures. First, is packet discarding, wherein a packet is discarded if all buffers are full. Second, is preallocating buffer space, an assimilation to virtual circuit thinking. Thirdly, separate restrictions are placed on the number of packets allowed in the network. Under the packet discarding methodology a station continues to transmit the same packet to another station and does not stop until the other station receives that packet and transmits an ACK. The routing algorithm described earlier handles this type of situation and thwarts congestion by rerouting the offered packet to other stations. Buffer allocation is a message priority scheme which proves to be best suited in either virtual circuit systems or file transfer modes; therefore, this method is not considered. This leaves the message restriction idea for implementation.

Restricting the number of packets allowed in the system can be accomplished in a number of ways. The implementation can be considered for the whole network, for a source destination pair, or for each station independent of all other stations. The first method is called Isarithmic control and is implemented through the issuance of permits. To transmit a packet a station must first obtain a permit. This concept does not

necessarily guarantee an equal distribution of permits throughout the network and requires a control station to handle the problem of lost or destroyed permits. Therefore, the best means of restricting the number of packets in the network must be either limiting the source-destination pair traffic (end-to-end control) or limiting the stations traffic (local control).

The mobile communication system under investigation requires that the stations relay all packets; therefore, it can be modeled as a series of tandem links. Pennotti and Schwartz [15],[17] used Fig. 5 to model end-to-end control.

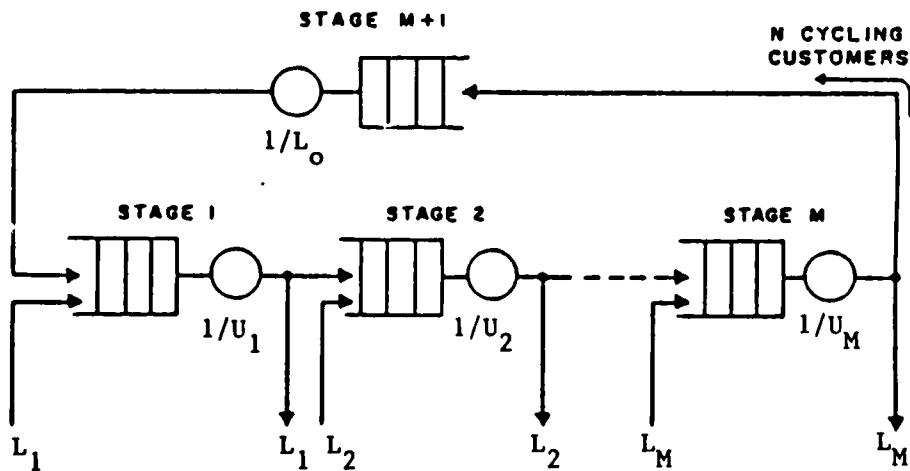


Fig. 5. End-to-end control model.

Using their equations for the expected number of link (n_i) and external (m_i) customers

$$E(n_i) = \frac{x_i}{Z_M^N} \cdot \frac{d}{dx_i} Z_M^N \quad \text{where } x_i = \frac{L_0}{U_i - L_i}$$

$$E(m_i) = \frac{L_i}{U_i - L_i} [1 + E(n_i)]$$

the end-to-end blocking probability is determined to be

$$P_B = 1 - \left(\frac{Z_M^{N-1}}{Z_M^N} \right)$$

where Z_M^N is calculated by [1]

$$Z = \sum_{\substack{\text{all partitions} \\ \text{of } n \leq N \text{ in } M}} \prod_{i=1}^M \left(\frac{L_0}{U_i - L_i} \right)^{n_i}$$

and the measure of congestion due to the presence of external customers is given as

$$C = \frac{1}{L} \sum_{i=1}^M [(U_i - L_i) E(m_i) - L_i]$$

or for link customers

$$C = \frac{1}{L} \sum_{i=1}^M L_i E(n_i)$$

Pennotti and Schwartz also investigated a local control option where each node would control the message flow. This investigation yielded similar results in the relative measures of congestion and blocking.

Both schemes try to give priority to packets already in the network. In an effort to lower the blocking probability the network suffers from a corresponding increase in packet delay time. This increase in congestion caused by the link customers is shown in Fig. 6 for both end-to-end and local control methods.

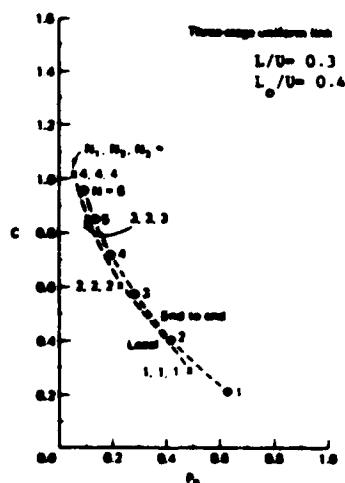


Fig. 6. Congestion versus blocking probability.

It is noted that the tandem link model accurately depicts either a one connected network or a mobile network in a flooding mode of operation due to their inherent means of packet deployment. But incorporating the proposed routing algorithm which extensively utilizes packet discarding, the calculated measure of congestion now becomes an upper bound on congestion for a mobile network with multiple paths to a destination.

It is therefore concluded that in a tandem link network both types of control schemes provide similar congestion control performance results with respect to link and external customer service making either a viable option. Since both low congestion and blocking probability are desired for implementation the end-to-end control option will be utilized due to the fact that its relative measure of congestion does not increase as quickly as that of the local control scheme with respect to small changes in the blocking probability.

CHAPTER 7

RESULTS

Selecting the following design specifications a network can now be formulated:

number of stations	10
mean arrival rate	all equal
mean service rate	all equal
mean initial arrival/service rate	.2
mean arrival/service rate	.3
channel bandwidth	100 KHz
maximum propagation delay	.0001

Before determining the frame size, the header information must be calculated. Utilizing the memory matrix concept in support of the route field the following header calculations are made

number of permutations	68,588,311
number of stack addresses	27
size of index matrix	6X6
number of route field bits	6
change field (20% threshold) bits	2
hop count field bits	4

With the three fields requiring only 12 bits a 1000 bit frame will be used. This frame requires 10 milliseconds to transmit (1 bit = 1 Hz).

The nonpersistent BTMA scheme described requires that additional parameters be selected. Choosing

probability of a correct detection	.5
fraction of the channel allocated to the busy tone channel	.01
detection time	.0005

the following plot in Fig. 7 is found.

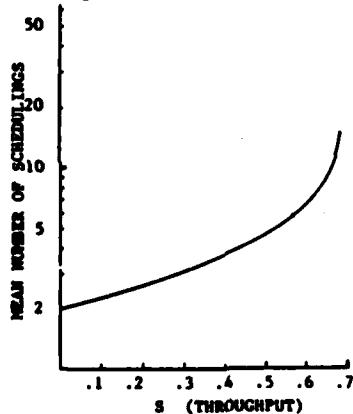


Fig. 7. BTMA mean number of schedulings and transmissions.

The maximum throughput is observed to be .68 with an average number of schedulings and transmissions (G/S) of 15.7.

Utilizing

$$t_d = (G/S)(1.5)/(U_1 C)$$

(assuming the retransmit delay is a uniform probability distribution between 0 and 3 packets long) this offers a mean packet transmission time of about .2355 seconds per hop under optimal conditions.

In calculating the blocking probability and the measure of congestion (the fractional increase in time delay) the following three cases will be considered (where R is the radius of transmission).

1. All stations are within reception range of each other (Fig. 8).
2. All stations are in two reception spheres with only one station connecting the two together (Fig. 9).
3. A one connected (worst case model) network (Fig. 4, on page 18).

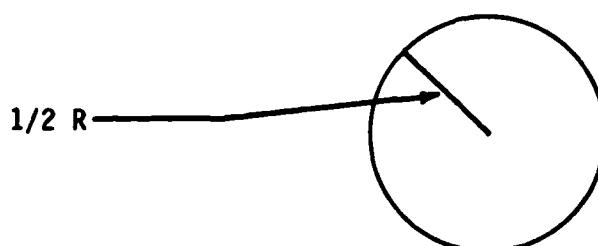


Fig. 8. Case 1 diagram.

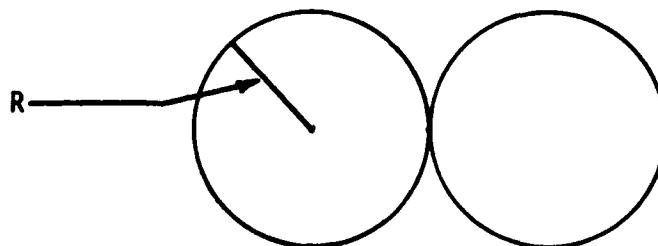


Fig. 9. Case 2 diagram.

Applying the congestion control equations, from page 26 (remembering that N is the number of packets allowed, and M is the number of stages), TABLE II is formulated.

Table II
Blocking Probability and Congestion Calculations

Case	Initial Arr/Svc Rate	Number of Messages	Blocking Probability	Congestion	Time Delay
1	.2	1	.222	.222	.4111
		5	1.36e-03	.3967	.4699
		10	2.59e-06	.3999	.4710
	.4	1	.364	.364	.4588
		5	.027	1.117	.7122
		10	1.59e-03	1.310	.7771
2	.2	1	.364	.1818	.9175
		5	5.85e-03	.3918	1.2001
		10	2.035e-05	.4000	1.211
	.4	1	.533	.2667	1.032
		5	.0767	.9755	1.986
		10	7.59e-03	1.273	2.385
3	.2	1	.72	.08	5.208
		5	.1455	.3095	11.461
		10	7.732e-03	.3935	13.750
	.4	1	.837	.0930	5.563
		5	.4289	.4437	15.120
		10	.1842	.8150	25.236

In calculating the total blocking probability, the transmitting station and the receiving queue(s) are independent of each other; therefore the probability of being blocked is

$$P_{B_T} = 1 - (P_A)(P_Q)$$

where P_A is the probability of accessing the medium (under optimal conditions $S = 68\%$), and P_Q is the probability that the queue is not full (1-the blocking probability from TABLE II).

TABLE II shows that the relative measure of congestion rises as the number of packets allowed in the system increases. This relative measure of congestion can be expressed in terms of time delay. The total time delay at a node can be expressed as

$$T_f = \text{service time} + \text{waiting time}$$

$$= \frac{A}{U_f C} + \frac{An}{U_f C}$$

where the time delay depends on four factors [5]; 1) the packets length ($1/U_f$), 2) the channel capacity (C), 3) the access and retransmit delay parameters (A), and 4) the expected number of packets in the queue (n).

The result, when $A=1$, is Little's formula

$$T_f = \frac{1}{U_f C(1-r)} \quad \text{substituting } r = \frac{L_f}{U_f C}$$

$$= \frac{1}{U_f C - L_f}$$

since $n=r/(1-r)$ for a M/M/1 queue.

Applying Little's result to Pennotti and Schwartz's formula for calculating congestion from page 26

$$C = \frac{1}{L} \sum_{i=1}^M [(U_i - L_i) E(m_i) - L_i]$$

$$= \sum_{i=1}^M \frac{L_i}{L} E(n_i)$$

and for a specific node

$$C = E(n_i)/M \quad (\text{where } M \text{ is the number of stages})$$

and using the following relations:

$$T_i = \frac{E(m_i)}{L_i}$$

where T_i is the average time delay incurred by external messages at node i ; and

$$\begin{aligned} T_{ni} &= \frac{E(m_{ni})}{L_i} \\ &= \frac{1}{U_i - L_i} \end{aligned}$$

where T_{ni} is the time delay when there are no link messages at node i; these substitutions reveal [15]

$$C = \sum_{i=1}^M \frac{L_i}{U_i} \frac{T_i - T_{ni}}{T_{ni}}$$

therefore

$$E(n_i) = \frac{T_i}{T_{ni}} - 1$$

Solving for T_i

$$T_i = T_{ni} (1+E(n_i)) = \frac{1}{U_i - L_i} (1+E(n_i))$$

substituting $E(n_i) = MC$ into the above equation results in

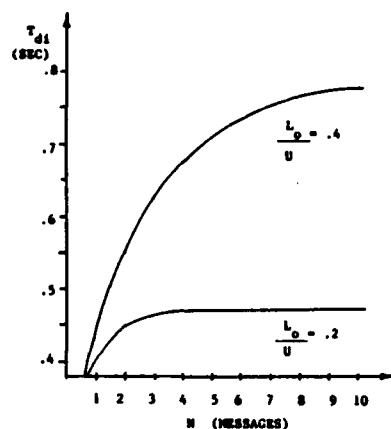
$$T_i = \frac{1}{U_i - L_i} (1+MC)$$

Calculating the time delay at node i, T_i must be multiplied by the number of attempts to access the medium and by the retransmission delay time (1.5 packets given earlier)

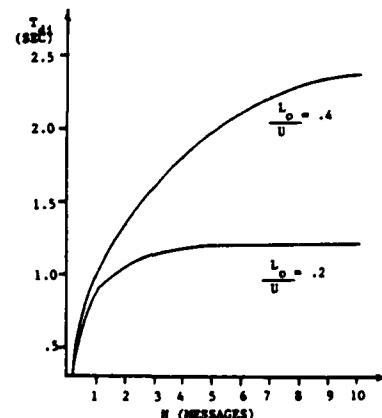
$$T_{di} = T_i (1.5)(G/S)$$

Consequently, the total delay time of the link is determined by multiplying T_{di} by the number of stages in the link.

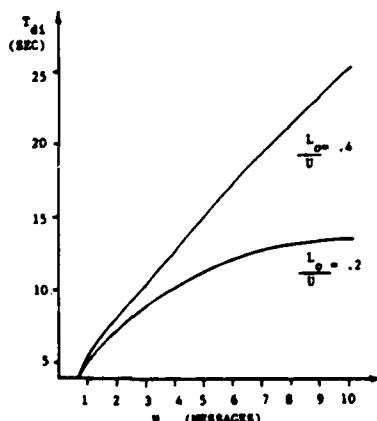
This equation coupled with the data from TABLE II, yields the time delay versus the number of messages for the three cases in question in Fig. 10.



a. Case 1 time delay plot.



b. Case 2 time delay plot.



c. Case 3 time delay plot.

Fig. 10. Time delay versus the number of messages.

In Case 1, all the stations are able to communicate with each other. Utilizing the .2 arrival/service rate and 1 message, a 47.1% blocking probability can be expected. This results if all the traffic is directed to one subscriber. Traffic not directed to the specific node utilizes the packet discarding option. The 47.1% can now be considered the upper bound and 32% (1-S) becomes the lower bound. The lower bound occurs as the number of packets in the link are allowed to increase; since the blocking probability at the queue is very small. Fig 10a shows that the .2 initial mean arrival/service rate can have a total packet delay time of .411 to .471 seconds (for 1 through 10 messages respectively). If the originating traffic doubles (the .4 condition of Fig. 10a) using 10 packets in the link, the blocking probability would be 32.1% and the expected time delay is .777 seconds.

Case 2 requires that the tandem links have a subscriber pass the traffic to the destination. Considering 10 packets in the network for the .2 initial mean arrival/service rate, a 1.211 second delay occurs, and if the arrival rate doubles (the .4 condition) a 2.385 second delay is expected. Similarly, in Case 3, the packet is passed through all the subscribers to its destination. Using the .2 initial mean arrival/service condition with 10 packets allowed a delay of 13.750 seconds would occur. Again, if the initial arrival rate were allowed to double (the .4 condition) the expected delay is 25.236 seconds. It is noted that the computer's processing time to determine the next hop is not included. The addition of this processing time will only add to the

delay. Realizing that the computer's clock operates from 3.5 to 8 MHz, this time is considered negligible and is therefore disregarded from these calculations.

Subsequently, using these parameters, the proposed mobile computer network shows workable results for a packet switched system. It is noted that Case 2 considered only one station passing traffic between the two spheres. Since no alternate routing capability was afforded the network, the upper bound on congestion had to be assumed. This measure of congestion can be reduced if two or more stations were to be placed between the two transmission spheres. This would provide alternate routing for the offered traffic. Again, this system is currently suited for packet deployment, but due to the fluctuations in delay caused by: the fluid topology and its respective congestion parameters; and the excessive time delays, preclude this systems' ability to implement voice capability.

CHAPTER 8

CONCLUSIONS

The mobile computer network proposed in this paper has the potential to be placed in service without modifications.

The nonpersistent BTMA method of seizing the channel yields a higher access rate than does either of the ALOHA methodologies, but there still exists, at best, a thirty two percent blocking probability associated with the channel access. This combined with the small blocking probabilities of the dependent stations calculated in the congestion section yield a higher probability of being blocked. It is noted that in the congestion section the blocking probability was inversely related to the level of congestion. Therefore, at the expense of a lower blocking probability the network accepts a higher level of possible congestion.

Since not too much is known about the movement of mobile subscribers in a network a highly adaptive routing algorithm is required. The routing algorithm described in this paper affords the network this adaptability, but requires a considerable amount of hardware and computer time to implement. In very large networks, the size of the header coupled with its corresponding memory support would appear to make this algorithm undesirable. The route field with its support memory, is attractive because: it channels the packets toward their destination in a totally mobile environment; it is highly adaptive in nature; it avoids

congested areas; it precludes excessive packet proliferation (compared to the standard flooding algorithms employed in broadcast networks); and it predetermines the packet's life span in the network.

The control of congestion in any switched network has already been emphasized in this paper. Since not a lot is known about the movement of stations in a mobile environment, the only available means of investigating congestion is to consider specific static cases. The tandem link model was used for just this purpose. The results showed that in trying to lower the blocking probability by allowing more packets to exist in the system, the level of congestion rose. This result is valid in a mobile network under the constraints of the standard flooding algorithm. But, the routing algorithm incorporated in this proposed network employs the flooding routine for only three purposes: first, for the initial source packet to locate the destination and to establish the route for the subsequent packets in the message; second it is used in single increments to detour blocked or congested areas; and third, it is used again in single increments to locate a station that has moved out of reception range. During subsequent packet transmissions the routing algorithm affords the stations the opportunity to maximize the packet discarding option. Considering this, under normal conditions the calculated values of congestion can now be considered an upper bound, and for a one connected network it is the measure of congestion.

On its face, this proposed mobile computer network satisfies the requirement that the terminals be able to communicate with each other and

that each station has the ability to double as a network relay. This network lends itself nicely to rapid deployment and emergency teams where there is either no time to commandeer a plethora of telephone lines for service or where there is no established landline network to "hook up" into. This network is obviously quite versatile and affords these teams the ability move freely about.

CHAPTER 9

SUGGESTED AREAS FOR INVESTIGATION

This network is very computer intensive and provides several areas for further investigation just from the two layers discussed in this paper. A possible area for investigation in the channel access realm would be that of determining an access method specifically tailored to the mobile network, that yields a high probability of seizing the channel. In the routing algorithm area further investigation may be pursued in changing the route field from a fixed length to a dynamically adjusted length (if the destination is in close proximity to the source.) The next logical step would then be to investigate the effects of variable size packets in the network with respect to user needs and system load. Further investigation is also needed in the areas of mobile subscriber movement and traffic flow patterns.

Outside the Data link and Network layers, further investigation should be conducted on the physical layer technology. The effects of noise, multipath and capture should be investigated. The use of directional antennas is another possible area to be considered. Efforts should be directed to the development of a new type of device that has three states versus the standard two used in switches today. All in all, this proposed network opens many possible avenues for future research.

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BIOGRAPHICAL SKETCH

Steven John Janis was born in New York, New York on November 7, 1956. Upon graduating from Hendrick Hudson High School in 1974, he entered the U.S. Army. He later went on to receive his Bachelor of Science degree in General Engineering from the United States Military Academy in 1979. He was commissioned a Second Lieutenant in the Signal Corps and has served in both command and staff capacities before the Department of the Army selected him to enter an advanced degree program. Currently, Captain Janis is pursuing a Master of Science Degree at Arizona State University in the field of Electrical Engineering.

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